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APPLICATION NO.	FILING DATE	FIRST NAMED INVENTOR	ATTORNEY DOCKET NO.	CONFIRMATION NO.
09/302,397	04/30/1999	KAZUNORI OZAWA	SON-0432	6830

30743 7590 01/28/2004

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EXAMINER

ARMSTRONG, ANGELA A

ART UNIT PAPER NUMBER

2654

DATE MAILED: 01/28/2004

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Please find below and/or attached an Office communication concerning this application or proceeding.

**Office Action Summary**

Application No.

09/302,397

Applicant(s)

OZAWA, KAZUNORI

Examiner

Angela A. Armstrong

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-- The MAILING DATE of this communication appears on the cover sheet with the correspondence address --

**Period for Reply**

A SHORTENED STATUTORY PERIOD FOR REPLY IS SET TO EXPIRE 3 MONTH(S) FROM THE MAILING DATE OF THIS COMMUNICATION.

- Extensions of time may be available under the provisions of 37 CFR 1.136(a). In no event, however, may a reply be timely filed after SIX (6) MONTHS from the mailing date of this communication.
- If the period for reply specified above is less than thirty (30) days, a reply within the statutory minimum of thirty (30) days will be considered timely.
- If NO period for reply is specified above, the maximum statutory period will apply and will expire SIX (6) MONTHS from the mailing date of this communication.
- Failure to reply within the set or extended period for reply will, by statute, cause the application to become ABANDONED (35 U.S.C. § 133).
- Any reply received by the Office later than three months after the mailing date of this communication, even if timely filed, may reduce any earned patent term adjustment. See 37 CFR 1.704(b).

**Status**

- 1) ☒ Responsive to communication(s) filed on 12 November 2003.
- 2a) ☒ This action is **FINAL**.                      2b) ☐ This action is non-final.
- 3) ☐ Since this application is in condition for allowance except for formal matters, prosecution as to the merits is closed in accordance with the practice under *Ex parte Quayle*, 1935 C.D. 11, 453 O.G. 213.

**Disposition of Claims**

- 4) ☒ Claim(s) 1-4 and 6-11 is/are pending in the application.
- 4a) Of the above claim(s) \_\_\_\_\_ is/are withdrawn from consideration.
- 5) ☐ Claim(s) \_\_\_\_\_ is/are allowed.
- 6) ☒ Claim(s) 1-4 and 6-11 is/are rejected.
- 7) ☐ Claim(s) \_\_\_\_\_ is/are objected to.
- 8) ☐ Claim(s) \_\_\_\_\_ are subject to restriction and/or election requirement.

**Application Papers**

- 9) ☐ The specification is objected to by the Examiner.
- 10) ☐ The drawing(s) filed on \_\_\_\_\_ is/are: a) ☐ accepted or b) ☐ objected to by the Examiner.
- Applicant may not request that any objection to the drawing(s) be held in abeyance. See 37 CFR 1.85(a).
- 11) ☐ The proposed drawing correction filed on \_\_\_\_\_ is: a) ☐ approved b) ☐ disapproved by the Examiner.
- If approved, corrected drawings are required in reply to this Office action.
- 12) ☐ The oath or declaration is objected to by the Examiner.

**Priority under 35 U.S.C. §§ 119 and 120**

- 13) ☐ Acknowledgment is made of a claim for foreign priority under 35 U.S.C. § 119(a)-(d) or (f).
- a) ☐ All   b) ☐ Some \* c) ☐ None of:
1. ☐ Certified copies of the priority documents have been received.
2. ☐ Certified copies of the priority documents have been received in Application No. \_\_\_\_\_.
3. ☐ Copies of the certified copies of the priority documents have been received in this National Stage application from the International Bureau (PCT Rule 17.2(a)).
- \* See the attached detailed Office action for a list of the certified copies not received.
- 14) ☐ Acknowledgment is made of a claim for domestic priority under 35 U.S.C. § 119(e) (to a provisional application).
- a) ☐ The translation of the foreign language provisional application has been received.
- 15) ☐ Acknowledgment is made of a claim for domestic priority under 35 U.S.C. §§ 120 and/or 121.

**Attachment(s)**

- 1) ☒ Notice of References Cited (PTO-892)                      4) ☐ Interview Summary (PTO-413) Paper No(s). \_\_\_\_\_
- 2) ☐ Notice of Draftsperson's Patent Drawing Review (PTO-948)                      5) ☐ Notice of Informal Patent Application (PTO-152)
- 3) ☐ Information Disclosure Statement(s) (PTO-1449) Paper No(s) \_\_\_\_\_                      6) ☐ Other: \_\_\_\_\_

### DETAILED ACTION

The following is a quotation of 35 U.S.C. 103(a) which forms the basis for all obviousness rejections set forth in this Office action:

(a) A patent may not be obtained though the invention is not identically disclosed or described as set forth in section 102 of this title, if the differences between the subject matter sought to be patented and the prior art are such that the subject matter as a whole would have been obvious at the time the invention was made to a person having ordinary skill in the art to which said subject matter pertains. Patentability shall not be negated by the manner in which the invention was made.

1. Claims 1-4 and 6-11 are rejected under 35 U.S.C. 103(a) as being unpatentable over Kleijn et al (US Patent No. 5,704,003) in view of Ozawa et al, "M-LCELP Speech Coding at 4KBPS", Acoustics, Speech, and Signal Processing, 1994. ICASSP-94, 1994 IEEE International Conference on, Volume: 1, 19-22 April 1994, Page(s): I/269 -I/272 vol.1.

2. Regarding claims 1 and 6, Kleijn et al teaches a spectrum parameter calculation section for receiving a speech signal, obtaining a spectrum parameter, and quantizing the spectrum parameter at col. 5, lines 66-67 and col. 6, lines 1-3.

Additionally, Kleijn teaches an adaptive codebook section for obtaining a delay and a gain and obtaining a residual by predicting a speech signal at col. 6, lines 52-62;

Kleijn also discloses a discrimination section for discriminating a voiced/unvoiced mode at col. 7 lines 10-26 and col. 5, lines 7-8;

Kleijn et al do not specifically teach that the discriminating a voiced/unvoiced mode is based on a past quantized gain of an adaptive codebook. However, determination of a voiced/unvoiced mode based on a past-quantized gain of an adaptive codebook was well known in the art.

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In a similar field of endeavor, Ozawa et al teaches a speech coding system (implementing encoder and decoder structures) which implements mode selection such that coding methods and codebooks are changed to improve coding efficiency as well as to reduce codebook size, in accord with each of 4 modes, for unvoiced and transition segments, and voiced segments. The mode is determined by comparing the average of the pitch prediction gain with each of three thresholds (page I-269, section 2).

Therefore, it would have been obvious to one of ordinary skill at the time of the invention to modify the coding system of Kleijn et al to implement discriminating a voiced/unvoiced mode is based on a past quantized gain, as taught by Ozawa et al, for the purpose of improving coding efficiency and reducing codebook size, as suggested by Ozawa et al.

Kleijn et al further teach, sound source quantization which has a codebook for representing a signal by combination of pulses and amplitudes and searches code vectors stored in the codebook and delays or shift amounts so as to output a combination of code vector and shift amount that minimizes distortion at col. 6, lines 21-61.

Kleijn does not specifically teach a multiplexer for the coder or a decoder scheme with a demultiplexer and sound source reconstruction. However, implementation of a multiplexer and providing an analogous decoding scheme for a specific coding system was well known.

Ozawa implements a M-LCELP encoder and decoder structure, which includes multiplexer on the encoder and demultiplexer with the decoder.

Therefore, it would have been obvious to one of ordinary skill at the time of invention to implement a multiplexer and a decoding scheme with the system of Kleijn et al for the purpose of providing high quality speech coding and decoding as suggested by Ozawa et al.

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Regarding claims 2 and 7, Kleijn discloses spectrum parameter calculation section for receiving a speech signal, obtaining a spectrum parameter, and quantizing the spectrum parameter at col. 5, lines 66-67 and col. 6, lines 1-3;

Additionally, Kleijn disclose an adaptive codebook section for obtaining a delay and a gain and obtaining a residual by predicting a speech signal at col. 6, lines 52-62 and a discrimination section for discriminating a voiced/unvoiced mode at col. 7 lines 10-26 and col. 5, lines 7-8;

Kleijn et al do not specifically teach that the discriminating a voiced/unvoiced mode is based on a past quantized gain of an adaptive codebook. However, determination of a voiced/unvoiced mode based on a past-quantized gain of an adaptive codebook was well known in the art.

In a similar field of endeavor, Ozawa et al teaches a speech coding system (implementing encoder and decoder structures) which implements mode selection such that coding methods and codebooks are changed to improve coding efficiency as well as to reduce codebook size, in accord with each of 4 modes, for unvoiced and transition segments, and voiced segments. The mode is determined by comparing the average of the pitch prediction gain with each of three thresholds (page I-269, section 2).

Therefore, it would have been obvious to one of ordinary skill at the time of the invention to modify the coding system of Kleijn et al to implement discriminating a voiced/unvoiced mode is based on a past quantized gain, as taught by Ozawa et al, for the purpose of improving coding efficiency and reducing codebook size, as suggested by Ozawa et al.

Kleijn et al further teach, sound source quantization which has a codebook for representing a signal by combination of a plurality of non-zero pulses and collectively quantizing amplitudes or polarities of the pulses based on an output from said discriminating section, and outputs a code vector that minimizes distortion relative to input speech by generating positions of the pulses according to a predetermined rule at col. 6, lines 21-61.

Kleijn does not specifically teach a multiplexer for the coder or a decoder scheme with a demultiplexer and sound source reconstruction. However, implementation of a multiplexer and providing an analogous decoding scheme for a specific coding system was well known.

Ozawa implements a M-LCELP encoder and decoder structure, which includes multiplexer on the encoder and demultiplexer with the decoder.

Therefore, it would have been obvious to one of ordinary skill at the time of invention to implement a multiplexer and a decoding scheme with the system of Kleijn et al for the purpose of providing high quality speech coding and decoding as suggested by Ozawa et al.

Regarding claims 3, 8 and 11, Kleijn discloses a spectrum parameter calculation section for receiving a speech signal, obtaining a spectrum parameter, and quantizing the spectrum parameter at col. 5, lines 66-67 and col. 6, lines 1-3;

Additionally, Kleijn discloses an adaptive codebook section for obtaining a delay and a gain and obtaining a residual by predicting a speech signal at col. 6, lines 52-62;

Kleijn discloses a discrimination section for discriminating a voiced/unvoiced mode at col. 7 lines 10-26 and col. 5, lines 7-8;

Kleijn et al do not specifically teach that the discriminating a voiced/unvoiced mode is based on a past quantized gain of an adaptive codebook. However, determination of a

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voiced/unvoiced mode based on a past-quantized gain of an adaptive codebook was well known in the art.

In a similar field of endeavor, Ozawa et al teaches a speech coding system (implementing encoder and decoder structures) which implements mode selection such that coding methods and codebooks are changed to improve coding efficiency as well as to reduce codebook size, in accord with each of 4 modes, for unvoiced and transition segments, and voiced segments. The mode is determined by comparing the average of the pitch prediction gain with each of three thresholds (page I-269, section 2).

Therefore, it would have been obvious to one of ordinary skill at the time of the invention to modify the coding system of Kleijn et al to implement discriminating a voiced/unvoiced mode is based on a past quantized gain, as taught by Ozawa et al, for the purpose of improving coding efficiency and reducing codebook size, as suggested by Ozawa et al.

Kleijn et al further teach, sound source quantization which has a codebook for representing a signal by combination of a plurality of non-zero pulses and collectively quantizing amplitudes or polarities of the pulses based on an output from the discriminating section, and a gain codebook for quantizing gains, and searches combinations of code vectors stored in said codebook, a plurality of shift amounts used to shift positions of the pulses, and gain code vectors stored in said gain codebook so as to output a combination of a code vector, shift amount, and gain code vector which minimizes distortion relative to input speech at col. 6, lines 21-61.

Kleijn does not specifically teach a multiplexer for the coder or a decoder scheme with a demultiplexer and sound source reconstruction. However, implementation of a multiplexer and providing an analogous decoding scheme for a specific coding system was well known.

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Ozawa implements a M-LCELP encoder and decoder structure, which includes multiplexer on the encoder and demultiplexer with the decoder.

Therefore, it would have been obvious to one of ordinary skill at the time of invention to implement a multiplexer and a decoding scheme with the system of Kleijn et al for the purpose of providing high quality speech coding and decoding as suggested by Ozawa et al.

Regarding claim 4, Kleijn teaches a spectrum parameter calculation section for receiving a speech signal, obtaining a spectrum parameter, and quantizing the spectrum parameter at col. 5, lines 66-67 and col. 6, lines 1-3;

Additionally, Kleijn teaches an adaptive codebook section for obtaining a delay and a gain and obtaining a residual by predicting a speech signal at col. 6, lines 52-62; Discrimination section for discriminating a voiced/unvoiced mode at col. 7 lines 10-26 and col. 5, lines 7-8;

Kleijn et al do not specifically teach that the discriminating a voiced/unvoiced mode is based on a past quantized gain of an adaptive codebook. However, determination of a voiced/unvoiced mode based on a past-quantized gain of an adaptive codebook was well known in the art.

In a similar field of endeavor, Ozawa et al teaches a speech coding system (implementing encoder and decoder structures) which implements mode selection such that coding methods and codebooks are changed to improve coding efficiency as well as to reduce codebook size, in accord with each of 4 modes, for unvoiced and transition segments, and voiced segments. The mode is determined by comparing the average of the pitch prediction gain with each of three thresholds (page I-269, section 2).



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Therefore, it would have been obvious to one of ordinary skill at the time of the invention to modify the coding system of Kleijn et al to implement discriminating a voiced/unvoiced mode is based on a past quantized gain, as taught by Ozawa et al, for the purpose of improving coding efficiency and reducing codebook size, as suggested by Ozawa et al.

Kleijn et al further teaches, sound source quantization which has a codebook for representing a signal by combination of a plurality of non-zero pulses and collectively quantizing amplitudes or polarities of the pulses based on an output from said discrimination section, and a gain codebook for quantizing gains, and searches combinations of code vectors stored in said codebook, a plurality of shift amounts used to shift positions of the pulses, and gain code vectors stored in said gain codebook so as to output a combination of a code vector, shift amount, and gain code vector which minimizes distortion relative to input speech at col. 6, lines 21-61.

Kleijn does not specifically teach a multiplexer for the coder or a decoder scheme with a demultiplexer and sound source reconstruction. However, implementation of a multiplexer and providing an analogous decoding scheme for a specific coding system was well known.

Ozawa implements a M-LCELP encoder and decoder structure, which includes multiplexer on the encoder and demultiplexer with the decoder.

Therefore, it would have been obvious to one of ordinary skill at the time of invention to implement a multiplexer and a decoding scheme with the system of Kleijn et al for the purpose of providing high quality speech coding and decoding as suggested by Ozawa et al.

Regarding claims 9-10, Kleijn and Ozawa et al teach everything as claimed in claim 8. Additionally, at col.7, lines 2-26, Kleijn discloses determination of the time shift amount is based on a value that minimizes a certain criteria, which reads on “sound source quantization uses a

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position generated according to a predetermined rule as a pulse position when mode discrimination indicates a predetermined mode.”

***Response to Arguments***

3. Applicant's arguments filed November 12, 2003 have been fully considered but they are not persuasive.

Applicant argues the modifications of Kleijn as proposed by the Examiner is not fairly taught by the references and it is not at all clear that such modifications would result in a working coder/decoder and that the Examiner has identified out of context features in Ozawa and say that they could be used to modify Kleijn.

In response to applicant's argument that there is no suggestion to combine the references, the examiner recognizes that obviousness can only be established by combining or modifying the teachings of the prior art to produce the claimed invention where there is some teaching, suggestion, or motivation to do so found either in the references themselves or in the knowledge generally available to one of ordinary skill in the art. See *In re Fine*, 837 F.2d 1071, 5 USPQ2d 1596 (Fed. Cir. 1988) and *In re Jones*, 958 F.2d 347, 21 USPQ2d 1941 (Fed. Cir. 1992). In this case, Ozawa et al teaches a speech coding system (implementing encoder and decoder structures) which implements a M-LCELP encoder and decoder structure, which includes multiplexer on the encoder and demultiplexer with the decoder and provides for mode selection such that coding methods and codebooks are changed to improve coding efficiency as well as to reduce codebook size. Further, the test for obviousness is not whether the features of a secondary reference may be bodily incorporated into the structure of the primary reference; nor

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is it that the claimed invention must be expressly suggested in any one or all of the references.

Rather, the test is what the combined teachings of the references would have suggested to those of ordinary skill in the art. See *In re Keller*, 642 F.2d 413, 208 USPQ 871 (CCPA 1981).

Regarding claims 9 and 10, Applicant argues the citation of Kleijn (col.7, lines 2-26) does not provide support for the claimed limitations. The Examiner disagrees and argues the teachings of Kleijn describe the criteria of delay and shifting of the pulse positions.

### *Conclusion*

4. **THIS ACTION IS MADE FINAL.** Applicant is reminded of the extension of time policy as set forth in 37 CFR 1.136(a).

A shortened statutory period for reply to this final action is set to expire THREE MONTHS from the mailing date of this action. In the event a first reply is filed within TWO MONTHS of the mailing date of this final action and the advisory action is not mailed until after the end of the THREE-MONTH shortened statutory period, then the shortened statutory period will expire on the date the advisory action is mailed, and any extension fee pursuant to 37 CFR 1.136(a) will be calculated from the mailing date of the advisory action. In no event, however, will the statutory period for reply expire later than SIX MONTHS from the mailing date of this final action.

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5. Any inquiry concerning this communication or earlier communications from the examiner should be directed to Angela A. Armstrong whose telephone number is 703-308-6258. The examiner can normally be reached on Monday-Thursday 7:30-5:00 PM.

If attempts to reach the examiner by telephone are unsuccessful, the examiner's supervisor, Richemond Dorvil can be reached on (703) 305-9645. The fax phone number for the organization where this application or proceeding is assigned is (703) 872-9306.

Any inquiry of a general nature or relating to the status of this application or proceeding should be directed to the receptionist whose telephone number is 703-306-0377.

Angela A. Armstrong  
Examiner  
Art Unit 2654

AAA  
January 25, 2004

  
RICHEMOND DORVIL  
SUPERVISORY PATENT EXAMINER